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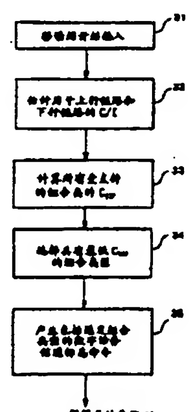
[54]发明名称 无线电电信网络中用于可变编码、调制及时隙分配的系统和方法

[57]摘要

本发明提供了一种用于动态地自适应一个时分多址联接（TDMA）蜂窝电信系统的用户比特率以获得在C/I条件的宽范围的最佳话音质量的系统和方法。该系统连续地监测上行链路（从一个移动站到其服务基站）和下行链路（从服务基站到该移动站）上的无线电信道质量，并动态地自适应于语音编码（21）、信道编码（22）、调制（23）和每次呼叫的可分配时隙数（27）的蜂窝系统的组合，以使所测定的条件下的话音质量最佳。不同的语音编码、信道编码、调制和可分配时隙数的组合方式以组合类型（1—5）为标识并定义了相应的费用函数。通过识别和选择具有所测定的无线电信道条件下的最低费用的费用函数，该系统提供了在系统设计限定范围内可获得的最大话音质量。



权　　利　　要　　求　　书

1. 一种用于动态地优化数字蜂窝无线电电信网络中的话音质量的系统，所述网络具有多个在设定的比特率下工作的用户比特率部件，并且所述网络利用多个无线电信道来载送呼叫，所述系统包括：

用于监视和测定每个所述无线电信道上的条件的装置；

用于估计每个所述无线电信道的当前无线电信道质量的装置；

用于改变所述多个用户比特率部件中的每一个的比特率的装置；以及

用于动态地控制所述用于改变比特率的装置的装置以便于为所述每个无线电信道上的呼叫提供最大可达到话音质量的装置。

2. 如权利要求1所述的用于动态地优化数字蜂窝无线电电信网络中的话音质量的系统，其特征在于所述用于监视和测定每个无线电信道上的条件的装置包括用于连续地监视和测量所述条件的装置。

3. 如权利要求2所述的用于动态地优化数字蜂窝无线电电信网络中的话音质量的系统，其特征在于，还包括用于监视和测量影响可达到话音质量的蜂窝网络条件的装置。

4. 如权利要求3所述的用于动态地优化数字蜂窝无线电电信网络中的话音质量的系统，其特征在于所述影响可达到话音质量的蜂窝网络条件包括：

移动站（MS）性能；

蜂窝网络性能；以及

资费。

5. 如权利要求3所述的用于动态地优化数字蜂窝无线电电信网络中的话音质量的系统，其特征在于所述多个用户比特率部件包括一个语音编码器、一个信道编码器、一个调制器、一个语音解码器、一个信道解码器和一个解调器。

6. 如权利要求5所述的用于动态地优化数字蜂窝无线电电信网络中的话音质量的系统，其特征在于所述用于连续地监视和测定每个所述无线电信道上的条件的装置包括用于连续地监视和测量误码率（BER）和信号强度（SS）的装置。

7. 如权利要求6所述的用于动态地优化数字蜂窝无线电电信网络

络中的话音质量的系统，其特征在于所述用于动态地控制所述用于改变比特率的装置的装置包括：

用于定义多种组合类型的装置，所述多种组合类型中的每一种包括一个确定的、用于多个用户比特率部件中的每一个的比特率；

用于定义多个费用函数的装置，每个所述费用函数对应于所述多种组合类型中的一种；以及

用于识别和选择在所述测定无线电信道条件下提供最低费用的费用函数的装置。

8. 如权利要求 7 所述的用于动态地优化数字蜂窝无线电电信网络中的话音质量的系统，其特征在于所述用于定义多个费用函数的装置包括：

用于定义随无线电信道质量变化的费用的装置；

用于定义随蜂窝网络应用变化的费用的装置；以及

用于将所述随无线电信道质量变化的费用和所述随蜂窝网络应用变化的费用相加以获得所述多种组合类型中的每一种的总费用函数的装置。

9. 如权利要求 8 所述的用于动态地优化数字蜂窝无线电电信网络中的话音质量的系统，其特征在于所述用于定义多个费用函数的装置包括用于将资费应用到所述每种组合类型的总费用函数上的装置，所述资费根据网络应用、无线电信道质量和对系统资源的要求来调整总费用函数。

10. 一种用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的系统，所述网络具有多个在设定的比特率下工作的用户比特率部件，并且所述网络利用多个时隙在每一无线电信道载送多个呼叫，所述系统包括：

用于监视和测定每个所述无线电信道上的条件的装置；

用于估计每个所述无线电信道的当前无线电信道质量的装置；

用于改变所述多个用户比特率部件中的每一个的比特率的装置；

用于向选定的呼叫分配时隙的装置；以及

用于动态地控制所述改变比特率的装置和所述用于分配时隙的装置的装置以便于为所述每个无线电信道上的呼叫提供最大可达到

话音质量的装置.

11. 如权利要求 10 所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的系统，其特征在于所述用于监视和测定每个无线电信道上的条件的装置包括用于连续地监视和测量所述条件的装置.

12. 如权利要求 11 所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的系统，还包括用于监视和测量影响可达到话音质量的蜂窝网络条件的装置.

13. 如权利要求 12 所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的系统，其特征在于所述影响可达到话音质量的蜂窝网络条件包括:

可用时隙;

移动站（MS）性能;

蜂窝网络性能; 以及

资费.

14. 如权利要求 12 所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的系统，其特征在于所述多个用户比特率部件包括一个语音编码器、一个信道编码器、一个调制器、一个语音解码器、一个信道解码器和一个解调器.

15. 如权利要求 14 所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的系统，其特征在于所述用于连续地监视和测定每个所述无线电信道上的条件的装置包括用于连续地监视和测量误码率（BER）和信号强度（SS）的装置.

16. 如权利要求 15 所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的系统，其特征在于所述用于动态地控制所述用于改变比特率的装置和所述用于分配时隙的装置的装置包括:

用于定义多种组合类型的装置，所述多种组合类型中的每一种包括:

一个为多个用户比特率部件中的每一个而设定的比特率; 以及

用于每次呼叫的时隙分配;

用于定义多个费用函数的装置，每个所述费用函数对应于所述多

种组合类型中的一种；以及

用于识别和选择在所述测定无线电信道条件下提供最低费用的费用函数的装置。

17. 如权利要求16所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的系统，其特征在于所述用于定义多个费用函数的装置包括：

用于定义随无线电信道质量变化的费用的装置；

用于定义随蜂窝网络应用变化的费用的装置；以及

用于将所述随无线电信道质量变化的费用和所述随蜂窝网络应用变化的费用相加以获得所述多种组合类型中的每一种的总费用函数的装置。

18. 如权利要求17所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的系统，其特征在于所述用于定义多个费用函数的装置包括用于将资费应用到所述每种组合类型的总费用函数上的装置，所述资费根据网络应用、无线电信道质量和对系统资源的要求来调整总费用函数。

19. 一种用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的方法，所述网络具有多个在设定的比特率下独立工作的用户比特率部件，并且所述网络利用多个时隙在每一无线电信道载送多个呼叫，所述方法包括步骤：

监视和测定每个所述无线电信道上的条件；

估计每个所述无线电信道的当前无线电信道质量；以及

动态地改变所述比特率和分配时隙，以便于为所述每个无线电信道上的呼叫提供最大可达到话音质量。

20. 如权利要求19所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的方法，其特征在于所述监视和测定每个无线电信道上的条件的步骤包括连续地监视和测量所述条件的步骤。

21. 如权利要求20所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的方法，还包括监视和测量影响可达到话音质量的蜂窝网络条件的步骤。

22. 如权利要求21所述的用于动态地优化时分多址联接（TDMA）

蜂窝无线电电信网络中的话音质量的方法，其特征在于所述监视和测量蜂窝网络条件的步骤包括监视和测量可用时隙、移动站（MS）性能、蜂窝网络性能、以及资费。

23. 如权利要求21所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的方法，其特征在于所述动态地改变多个用户比特率部件中的每一个的比特率的步骤包括改变一个语音编码器、一个信道编码器、一个调制器、一个语音解码器、一个信道解码器和一个解调器的比特率。

24. 如权利要求23所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的方法，其特征在于所述连续地监视和测定每个所述无线电信道上的条件的步骤包括连续地监视和测量误码率（BER）和信号强度（SS）。

25. 如权利要求24所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的方法，其特征在于所述动态地改变所述比特率和分配时隙的步骤包括以下步骤：

定义多种组合类型，所述定义步骤进一步包括：

为多个用户比特率部件中的每一个设定一个比特率；以及

为每次呼叫分配时隙数；

定义多个费用函数，每个所述费用函数对应于所述多种组合类型中的一种；以及

识别和选择在所述测定无线电信道条件下提供最低费用的费用函数。

26. 如权利要求25所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的方法，其特征在于所述定义多个费用函数的步骤包括：

定义随无线电信道质量变化的费用；

定义随蜂窝网络应用变化的费用；以及

将所述随无线电信道质量变化的费用和所述随蜂窝网络应用变化的费用相加以获得所述多种组合类型中的每一种的总费用函数。

27. 如权利要求26所述的用于动态地优化时分多址联接（TDMA）蜂窝无线电电信网络中的话音质量的方法，其特征在于所述定义多个费用函数的步骤包括将资费应用到所述每种组合类型的总费用函数

上，所述资费根据网络应用、无线电信道质量和对系统资源的要求来调整总费用函数。

说　明　书

无线电电信网络中用于可变编码、调制及时隙分配的系统和方法

本发明涉及无线电通信系统，特别涉及一种通过可变编码、调制及时隙分配来改进声音质量和无线电信道质量的系统和方法.

在现代蜂窝式电信系统中，覆盖的地理区域可被分成多个连续的无线电覆盖区域或网孔，其中的每一个都通过一个基站进行服务. 如现有技术中所公知的，每个基站包括一个发射机、接收机及一个基站控制器. 通过通信线路将一个移动式服务交换中心（MSC）连接到每个基站以及公共交换电话网（PSTN）或一个类似的、包括综合业务数字网络（ISDN）功能的固定网络上. 同样，关于蜂窝式无线电系统中包括多于一个MSC且通过电缆或无线电链路将附加的每个MSC连接到不同的基站组和其它MSC上也是公知的. 一个移动站可以在服务区周围自由漫游. 当移动站在系统的服务区周围漫游时，它们从一个网孔切换到另一网孔以使得在服务中没有延误.

每个网孔分配有多个声音或话音信道以及至少一个接入或控制信道. 该控制信道用于通过从那些装置发送和接收的被称为消息的信息来控制或监督移动终端的操作. 按照行业建立的空中接口标准（如AMPS 和 EIA/TIA 553，用于模拟蜂窝操作的标准，和/或 D-AMPS, EIA/TIA627 及 TIA IS136，用于数字蜂窝操作的标准）在一个蜂窝无线电系统中发送控制和管理消息，上述所有标准在此被引用以作为参考. 而这些标准负责管理北美地区的操作，相似的标准负责管理世界其他地理地区的操作，这些已为本领域技术人员所公知.

通过消息而在基站与移动终端之间交换的信息可包括呼入信号、呼出信号、寻呼信号、寻呼响应信号、位置登记信号、话音信道分配、维护指令和当移动终端移动到一个网孔的无线电覆盖区之外而进入另一网孔的无线电覆盖区中时的越区切换指令，以及诸如主叫方号码、时间信息之类的其它附加信息项. 控制或话音信道可根据行业标准以模拟或数字模式或两者的组合模式来操作. 通过利用在此作为参考的系统内部条约 IS-41，提供在不同的蜂窝电信系统和不同的MSC之间的综合服务.

如今使用中的移动站的增长量产生了在蜂窝电信系统中需要更

多话音信道的要求．随着在工作于同一频率时邻近或密集网孔中的移动站之间干扰的增强，基站已变得更加密集．此外，由于分配给蜂窝电信系统的频谱是有限的，从而导致了信道频率随来自其他信道的干扰的增强而更加密集．现在已经开发了诸如信号的时分多路复用和码分多路复用之类的数字技术以便于从一个给定的频谱获得更有用的信道．

这些技术仍保留着对减少干扰，特别是增大载波信号强度与干扰强度的比值（即载波/干扰（C/I）比）的需求．这里将 C/I 定义为总载波/干扰比，其中干扰包括来自其它移动站的干扰以及噪声（接收机产生的噪声和热噪声）．

在蜂窝无线电系统中，用户比特率是一种有限的资源．对一个具有一给定的用户比特率的系统来说，在无差错条件（高 C/I 比）下的话音质量与针对较差无线电信道质量（低 C/I 比）的系统耐久性之间需要有所折衷．

在以其它特征为代价的情况下，系统可以对话音质量或耐久性赋予优先权．例如，一个具有将优先权赋予了无差错条件下的话音质量的用户比特率的系统在高 C/I 等级下运行良好，但是与一个具有将优先权赋予耐久性的用户比特率的系统相比，在低 C/I 等级下则缺乏稳定性．换言之，第一个系统的话音质量随着 C/I 等级的降低而恶化得更快．另外，一个具有将优先权赋予耐久性的用户比特率的系统对低 C/I 等级来说更稳定，但是不能象一个具有将优先权赋予话音质量的用户比特率的系统一样在高 C/I 等级下运行良好．换言之，耐久系统的话音质量随着 C/I 等级的降低而恶化得较慢，但不具有象较好的无线电条件下的话音质量一样好的话音质量．

通过用于语音编码、信道编码、调制和用于一时分多址联接（TDMA）系统的技术的一种选定组合来确定一个蜂窝无线电系统中的总用户比特率，每个网孔可分配的时隙数在空中接口标准中规定．通过这个空中接口标准（如 IS-136）来确定上述技术的一种固定组合．但是由于对可实现的话音质量的限制，因而使得这种规定的组合具有缺陷，这种缺陷是由于在不适当的无线电信道质量条件下使用规定组合而引起的．每种规定的组合最佳地适用于无线电信道质量的一个特定等级（C/I 比），因此在 C/I 比较高时损失话音质量，并/或在 C/I

比较低时损失耐久性。现如今的蜂窝空中接口标准规定了或者提供高 C/I 条件下的高话音质量或者提供耐久性的这个固定组合。用于高 C/I条件下高话音质量的组合产生了一个耐久性较小并且话音质量在低 C/I 条件下难以被接受的系统。用于耐久性的组合损失了高 C/I 条件的高话音质量，而换来了在低C/I 条件下可接受的话音质量，尽管可能只在有限次数的 C/I 条件较低的情况下需要该耐久性。

尽管现有技术中还没有解决上述所公开的缺陷的方法，但存在一些讨论主题与此处所讨论的主题相关的现有技术参考文献。这些现有技术参考文献是授予 Freeburg 等人的专利号为 5，134，615 的美国专利文献和作者为 J. Woodard 和 L. Hanzo 的题为"一种基于二元代数 CELP 的话音收发信机"的 IEEE 文章。下面简要介绍一下每篇参考文献。

授予 Freeburg 等人（Freeburg）的专利号为 5，134，615 的美国专利文献公开了一种用于与具有不同通信协议并包括不同的可用时隙的设备进行通信的选择频率和时隙分配的方法。它采用了一个可自适应的时隙选择器，同时允许与利用其它协议的设备进行通信。但是，Freeburg 只在提供与利用不同的空中接口协议的设备进行通信的上下文情况下对时隙分配进行寻址。Freeburg 没有以任何方式提出一种在 C/I 条件的宽范围下的数字蜂窝系统中获得改进的话音质量的方法。本发明动态地自适应语音编码、信道编码、调制和每次呼叫的可分配时隙数的蜂窝系统的组合，以获得当前测定的 C/I 条件下的最佳话音质量。

作者为 J. Woodard 和 L. Hanzo 的题为"一种基于二元代数 CELP 的话音收发信机"的 IEEE 文章（Woodard）公开了一种利用了两种所谓低质量模式和高质量模式的语音编码、信道编码和调制的组合。但是，Woodard 没有提出用于不同模式的质量驱动或容量驱动选择的处理方法。Woodard 没有以任何方式提出一种能够动态地自适应语音编码、信道编码、调制和每次呼叫的可分配时隙数的蜂窝系统的组合，以获得在 C/I 条件的宽范围下的最佳话音质量的系统。

考察了上述的各篇参考文献之后，没有得到任何对于如本发明所公开和要求获得保护的系统和方法的提示。

一种用于动态地自适应语音编码、信道编码、调制和每次呼叫的

可分配时隙数的蜂窝系统的组合，以获得超出宽范围的 C/I 条件的最佳话音质量的系统和方法将会具备显著的优点。本发明提供了这样一种系统和方法。

本发明提供了一种用于动态地自适应一个时分多址联接（TDMA）蜂窝电信系统的用户比特率以获得在 C/I 条件的宽范围下的最佳话音质量的系统和方法。该系统连续地监测上行链路（从一个移动站到其服务基站）和下行链路（从服务基站到该移动站）上的无线电信道质量，并动态地自适应语音编码、信道编码、调制和每次呼叫的可分配时隙数的蜂窝系统的组合，以使所测定的条件下的话音质量最佳。系统的语音编码、信道编码、调制和可分配时隙数的不同的组合方式以组合类型为标识。此外，还应介绍一下费用函数，并且通过识别和选择具有测定无线电信道条件下的最低费用的费用函数，该系统提供了在系统设计限定范围内可获得的最大话音质量。

在另一方面，本发明提供了一种用于动态地优化一个数字蜂窝无线电通信网络中的话音质量的系统，该网络具有多个在设定的比特率下工作的用户比特率部件。该网络利用多个无线电信道来载送呼叫。为了提供用于每个无线电信道上的呼叫的最大话音质量，该系统包括用于监视和测定每个无线电信道上的条件的装置、用于估计每个无线电信道的当前无线电信道质量的装置、用于改变多个用户比特率部件中的每一个的比特率的装置、以及用于动态地控制所述用于改变比特率的装置的装置。

在另一方面，本发明提供了一种用于动态地优化一个时分多址联接（TDMA）蜂窝无线电通信网络中的话音质量的方法，该网络具有多个在设定的比特率下工作的用户比特率部件。该网络利用多个时隙来载送每个无线电信道上的多个呼叫。该方法以监视和测定每个无线电信道上的条件并估计每个无线电信道的当前无线电信道质量为开始。然后该方法根据估计出的无线电信道质量而动态地改变比特率并分配时隙，从而提供用于每个无线电信道上的呼叫的最大话音质量。

通过参照附图及其说明，将更利用理解本发明并且对于本领域技术人员来说，本发明的目的和有益效果将更为明显，其中：

图 1（现有技术）是一个服务基站的移动站信号强度随移动站距基站的距离而变化的示图；

图2（现有技术）是一个用于说明蜂窝电信系统中的移动站（MS）百分比的典型累积分布函数图，该系统在一给定时间具有低于相应等级的载波信号强度与干扰比（C/I）；

图3（现有技术）是一个对总用户比特率产生影响的蜂窝电信系统中的基站和移动站的各部件简化方框图；

图4（现有技术）是话音质量随用于图3所示的部件的三种典型组合示例的载波信号与干扰比（C/I）而变化的示图；

图5是话音质量随载波信号与干扰比（C/I）而变化的示图，用于说明在不同的C/I比等级下通过自适应选择图3所示的部件的最佳组合对话音质量产生的影响；

图6是用于说明在呼叫建立过程中动态选择组合类型时由本发明的控制程序执行的功能的流程图；

图7A和7B是用于说明在呼叫进程过程中动态选择组合类型时由本发明的控制程序执行的功能的流程图；

图8是用于说明在不同的无线电干扰和蜂窝系统容量条件下的五种典型组合类型可达到的话音质量等级的立体图；

图9是话音质量随表II中的五种典型组合类型的无线电信道质量（RCQ）或C/I而变化的示图；

图10是费用与本发明的控制程序使用的C/I比的关系曲线，用以说明不同资费（TARIFFS）下的曲线；

图10是费用（C_{RCQ}）随表II中的五种典型组合类型的无线电信道质量（RCQ）或C/I而变化的示图；

图11是费用（C_{SU}）随系统应用（SU）而变化的示图；

图12是总费用（C_{TOT}）随无线电信道质量和系统应用而变化的示图；以及

图13是组合类型I的总费用（C_{TOT}）曲线图，用以说明不同的“资费”在总费用曲线上的应用。

图1是一个服务基站的移动站信号强度随移动站距基站的距离而变化的示图。当前系统测量在基站的移动站信号强度，并随着一个移动站相对其服务基站移远时，所测量的信号强度减弱。当该信号强度达到一个最小可接受等级11时，移动站的输出功率突然增大，从而将基站所接收的信号强度增至一个中间级12。该中间级12用以提供

一个可接受信号强度和可接受干扰等级给其它移动站。随着移动站继续增大其距基站的距离，信号强度再次减至最小可接受等级，且移动站的输出功率再次突然增大。连续出现此过程直至移动站工作于其最大输出能力下为止。如果信号强度降至最小可接受等级，那么如果可能的话，移动站被越区切换到另一个网孔或该呼叫下线。

图 2 是一个用于说明蜂窝电信系统中的移动站（MS）百分比的典型累积分布函数图，该系统在一给定时间具有低于相应等级的载波信号强度与干扰比（C/I）。如仅作为示例的图 2 所示，100%的移动站的 C/I 比低于或等于 50dB。当其近似于 80%时，C/I 比低于或等于 30dB。同样，在几乎没有移动站时，则 C/I 太低以致于不能保持呼叫。C/I 比通常被认为是代表一个给定的蜂窝无线电信道上的话音质量的测量值，比值越大则提供的话音质量越好。实际上 C/I 测量值还包括一个噪音因素，但在干扰受限制的环境中，与干扰的影响相比，噪音对话音质量的影响就可以忽略不计了。

图 3 是一个蜂窝电信系统中一个基站和一个移动站的对可达到的话音质量产生影响的部件简图。在现有的蜂窝电信系统中，通过组合发射时的语音编码器 21、信道编码器 22 和调制器 23 的比特率；接收时的解调器 24、信道解码器 25 和语音解码器 26 的比特率；以及对一个时分多址联接系统（TDMA）来说还有空中接口 27 中的每次呼叫可分配时隙数，从而来确定总比特率。上述每种部件的可允许的比特率由现有的电信标准所规定。比特率具有大量的可能组合，可由操作员进行选择以优先化话音质量或系统耐久性。

图 4 是话音质量随用于图 3 所示的三种典型组合示例的载波信号与干扰比（C/I）而变化的示图。虽然为简化起见只以三种组合进行说明，但应理解这只是举例，而实际使用中还可以利用更多组合。

组合类型 A、B 和 C 分别表示为说明用于不同的 C/I 比的每种组合的可获得的话音质量的曲线。组合类型 A 提供了三种用户比特率下的最佳话音质量，而其耐久性最差，它只适用于较高的 C/I 等级。在组合类型 A 中，话音质量随 C/I 降低而迅速衰减到一个不可接受的等级。反之，组合类型 C 是最耐久的。因此，随着 C/I 的降低，组合类型 C 下的话音质量衰减得非常慢并且组合类型 C 提供了低 C/I 等级下的最佳话音质量。但是，组合类型 C 牺牲了高 C/I 等级下的话音质量，

此时它所达到的话音质量是这三种组合示例中最差的。

组合类型B提供了组合类型A在高C/I区域的高话音质量性能与组合类型C的耐久性之间的一种折衷方式。与组合类型A或C相比，组合类型B提供了在C/I的中间范围的较高话音质量。在高C/I等级，组合类型B提供了高于组合类型C但低于组合类型A的话音质量。在低C/I区域，组合类型B提供了高于组合类型A但低于组合类型C的话音质量。类似于组合类型B的组合经常被蜂窝空中接口所利用，因为这些组合几乎在整个C/I范围提供了中等性能。

图5是话音质量随载波信号与干扰比（C/I）而变化的示图，用于说明在不同的C/I比等级下自适应选择图3所示的部件的最佳组合对话音质量产生的影响。自适应选择提供了一个具有图5中黑体曲线所示的可达到的话音质量的蜂窝无线电系统。在本发明的系统中，瞬时无线电信道质量（即C/I比）被连续地监测。对于上述的每种组合类型A、B和C来说，对应于测量的C/I比的话音质量和所需的耐久性水平是已知的。通过选择在测定的C/I等级下组合类型A、B和C中的哪种组合类型给出了对应于所需耐久性的最高话音质量，从而使得该系统动态地响应所测量的C/I。这样，利用图4和图5所示的曲线，系统在高C/I等级使用组合类型A，在C/I的中间级使用组合类型B，并在低C/I等级使用组合类型C。因此，从动态上来看，话音质量总是最高的。

本发明包括一个用于对一个给定的C/I等级选择最佳组合的控制算法。该控制算法的基础是提供了简单而稳定的决策方法的“费用函数”。不同的组合之间的转换可由蜂窝系统或移动站控制。

北美的一种当前TDMA标准（IS－136）规定了一种三时隙结构，即将每三个时隙分配给一个特定用户。图4和图5的话音质量曲线说明了在对于每个用户只使用三个时隙中的一个时隙时可达到的话音质量。其它的时隙可被分配给一个用户，但是这种分配由于减少了每个频率上的用户数因而有害地影响到系统容量。但是，对那个特定用户来说，将其它时隙分配给一个用户无疑增大了带宽并提高了话音质量。因此，在系统容量不成问题时，就希望在话务密度低的时期将其它时隙分配给每个用户。因此，根据当前负载和用户类别，可通过利用“资费”（费用函数集）来强化本发明的控制程序。这就为蜂窝系统操作

员提供了在话音质量与系统容量之间进行折衷选择的能力，或向希望为附加性能支付佣金的用户提供另外的带宽。

图 6 是用于说明在动态选择一种呼叫建立过程中的组合类型时由本发明的控制程序执行的功能的流程图。该程序由步骤 31 开始，最初将移动站接入蜂窝电信网络。然后程序移到步骤 32 并根据空闲信道测量，为上行链路信号（从移动站到基站）和下行链路信号（从基站到移动站）估计无线电信道质量（RCQ）（如 C/I）。在步骤 33，该程序计算由蜂窝网络和移动站支持的所有用户比特率组合类型的总费用（C_{TOT}）。在步骤 34，该程序选择具有用于上行链路和下行链路传输的最低总费用的组合类型。然后在步骤 35，该程序产生一个数字话务信道（DTC）标志命令，该命令包括关于用于上行链路和下行链路传输的组合类型信息。然后该程序转到图 7 的步骤 41。

图 7A 和 7B 是用于说明在动态选择一种呼叫进程过程中的组合类型时由本发明的控制程序执行的功能的流程图。在步骤 41，该程序连续地监测该无线电信道并估计用于上行链路和下行链路的无线电信道质量（C/I）。在步骤 42，该程序计算由蜂窝网络和移动站支持的所有用户比特率组合类型的总费用（C_{TOT}）。在步骤 43，该程序选择具有用于上行链路和下行链路传输的最低总费用的组合类型。在步骤 44，判断具有最低 C_{TOT} 的组合类型当前是否正在被上行链路和下行链路所使用。如果已判定具有最低 C_{TOT} 的组合类型当前正在被上行链路和下行链路所使用，则不进行任何操作，该程序返回步骤 41 并继续监测无线电信道和估计无线电信道质量。

但是，如果在步骤 44，判定具有最低 C_{TOT} 的组合类型当前没有被上行链路和下行链路所使用，则该程序执行将上行链路和下行链路改为一种新的用户比特率组合类型的功能。为完成此功能，该程序首先移到步骤 45 并判断具有最低 C_{TOT} 的组合类型是否需要改变时隙分配。如果不需要改变时隙分配，则程序移到图 7B 的步骤 46 并向移动站发送一个物理层控制消息，该消息包括有关用于上行链路和下行链路的新组合类型的信息。当该程序在步骤 47 接收到一个物理层控制确认消息时，向新组合类型的转换就完成了。该程序返回图 7A 的步骤 41 并继续监测无线电信道和估计无线电信道质量。

但是，如果在步骤 45，判断出具有最低 C_{TOT} 的组合类型需要改变

时隙分配，则启动一个越区切换。该程序首先移到图 7B 的步骤 48 并占用一个新的数字话务信道。然后该程序移到步骤 49 并向移动站发送一个越区切换消息，该消息包括有关用于上行链路和下行链路的新组合类型（包括新的时隙分配）的信息。当该程序在步骤 50 接收到一个越区切换确认消息时，向新组合类型的转换就完成了。该程序返回图 7A 的步骤 41 并继续监测无线电信道和估计无线电信道质量。

该控制程序连续地监视和测量用于确定上行链路和下行链路的无线电信道质量（RCQ）的无线电信道条件，以及可能对可达到的话音质量产生影响的其它蜂窝网络条件。这些条件包括，例如：

<u>无线电信道条件：</u>

误码率（BER）-上行链路；

误码率（BER）-下行链路；

信号强度（SS）-上行链路；以及

信号强度（SS）-下行链路。

<u>蜂窝网络条件：</u>

可用时隙；

移动站（MS）性能；

蜂窝系统性能；以及

资费。

控制程序监测这些条件并根据其测量值来优化每个单独呼叫的话音质量以在该蜂窝系统的给定资源（如时隙、MS 性能等）范围内达到最佳的话音质量。引入费用函数是为了在系统容量和话音质量之间提供灵活的折衷。上述测定的参数被输入到控制程序中，于是该程序利用费用函数选择使总费用最低的组合类型。

控制程序依据误码率（BER）估计值（上行链路和下行链路）和信号强度（SS）估计值（上行链路和下行链路）来估计当前的无线电信道质量（C/I）。控制程序可以利用类似于下述表 I 的查阅表来将 BER 转成 C/I。

表 I

上行		下行	
BER (%)	C/I (dB)	BER (%)	C/I (dB)
10	7	10	10
5	9	5	13
3	11	3	17

下面描述一个具体的示例来说明本发明的一种典型实施方式。在这个示例中，该蜂窝网络共有五（5）种可用组合类型。这五种组合类型的定义如下面表 II 所示：

表 II

类型	时隙[a]	调制	话音编码器算法，速率（kbps）		总数据率[b]（kbps）
1	5	8PSK	ADPCM	32	56.75
2	5	π/4 QPSK	LDCELP	16	37.8
3	2	8PSK	LDCELP	16	19.5
4[c]	2	π/4 QPSK	VSELP	7.95	13
5	2	π/4 QPSK	EVCELP[d]	4.0	13

[a] 每 40 ms 中 IS-136 空中接口时隙数(其总数为 6)

[b] 包括话音信号编码器费率和前向纠错（FEC）编码 [c] IS-136 全费率

[d] 增强的 VSELP

表中所用的缩写展开如下：

PSK	相移键控
QPSK	正交移相键控
ADPCM	自适应差分脉码调制
LDCELP	低延迟受激码线性预测编码
VSELP	受激向量和线性预测编码

组合类型 1 和 2 最适用于室内/局内应用，它在每个频率上的系统容量的困难较少，因为可通过实现微网孔来获得所需的容量。组合类型 3、4 和 5 最适用于室外/广域应用，这里需要每个网孔/频率上

的最大容量.

图8是用于说明在不同的无线电干扰和蜂窝系统容量条件下的五种典型组合类型可达到的话音质量等级的立体图. 组合类型1和3在较好的无线电条件(高C/I比)和不同的蜂窝系统容量等级下提供了非常好的话音质量. 组合类型2、4和5在降低后的无线电条件(低C/I比)的各种等级和不同的蜂窝系统容量等级下提供了最佳话音质量.

图9是话音质量随表II中的五种典型组合类型的无线电信道质量(RCQ)或C/I而变化的示图. 每条曲线表示一种组合类型. 从图9中可以看到组合类型2和5是最稳定的, 它给出条件降级的、但在较低的C/I等级下的可接受的话音质量. 相反地, 曲线1和3在高C/I等级提供了好的话音质量, 但在较低的C/I等级则迅速降低到不可接受的话音质量.

图10是费用(C_{RCQ})随表II中的五种典型组合类型的无线电信道质量(RCQ)或C/I而变化的示图. 每条曲线表示一种组合类型. 这些曲线表明与在较高的C/I等级达到可接受的话音质量相比, 在较低的C/I等级达到可接受的话音质量则付出的花费更大. 这个事实导致五种组合类型的费用曲线基本上与图9所示的话音质量曲线相反. 因此, 通过选定一给定无线电信道质量(C/I)下的最低费用曲线, 从而也就选择了提供最佳话音质量的组合类型.

图11是费用(C_{NU})随蜂窝网络应用(NU)而变化的示图. 这些曲线分别表示各种组合类型. 图11中的曲线表明在高级系统应用时期将其它的时隙分配给每个用户要付出更大的花费. 这是因为由于需要服务于更多用户而导致几乎没有可用时隙来提高话音质量.

图12是总费用(C_{TOT})随无线电信道质量和系统应用而变化的示图. 每种组合类型的总费用(C_{TOT})是该组合的C_{RCQ}和C_{NU}之和. 如上所述, 控制程序连续地监测无线电信道质量和系统应用, 并选择最低的总费用曲线. 这导致了在蜂窝系统资源的限定范围内选择能提供最佳话音质量的组合类型.

图13是组合类型I的总费用(C_{TOT})曲线图, 用以说明不同的资费在总费用曲线上的应用. 对具有不同优先权等级的用户组来说, 资费为蜂窝系统操作员提供了修整系统应用的能力. 达到这一等级的话

音质量可能需要接入一种利用附加时隙的组合类型。通过征收基于总费用函数的资费，系统操作员可以向那些愿意为获得附加时隙而支付加价的用户提供这种服务。操作员还能控制资费的多少，因此当可用时隙不足时，在高的系统应用时期中则具有更高的资费。从而资费根据网络应用、无线电信道质量和对系统资源的要求来调整总费用函数。

我们相信从上面的描述中，本发明的操作和结构将是显而易见的。但应当理解在不偏离本发明的发明构思和下述权利要求所限定的范围的前提下，对上面所示和所描述的方法、装置和系统的最佳实施方式作出各种变化和修改是可能的。

说 明 书 附 图

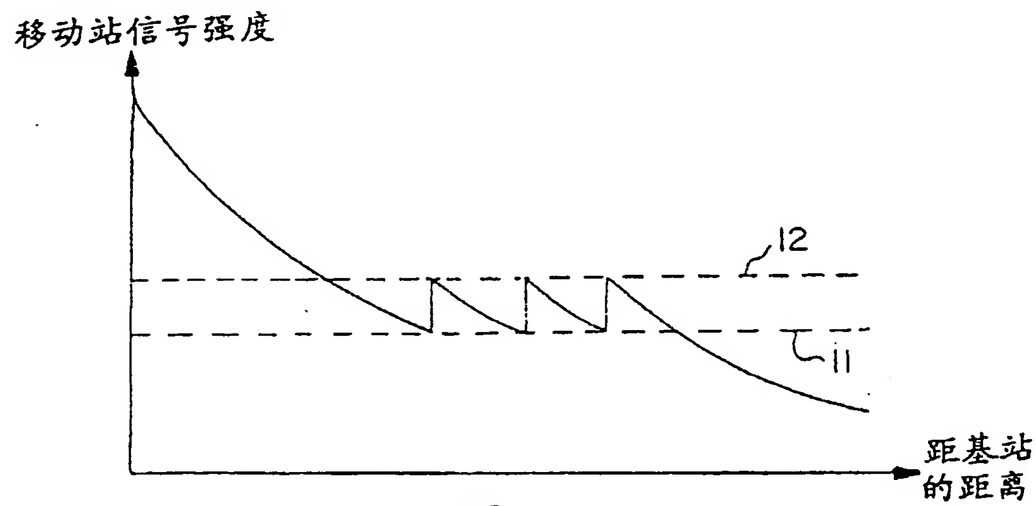


图 1

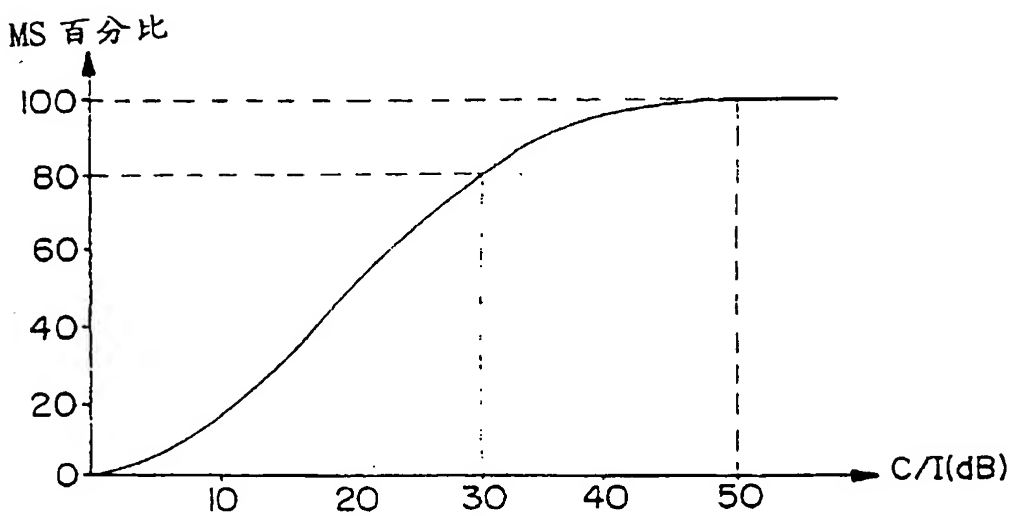


图 2

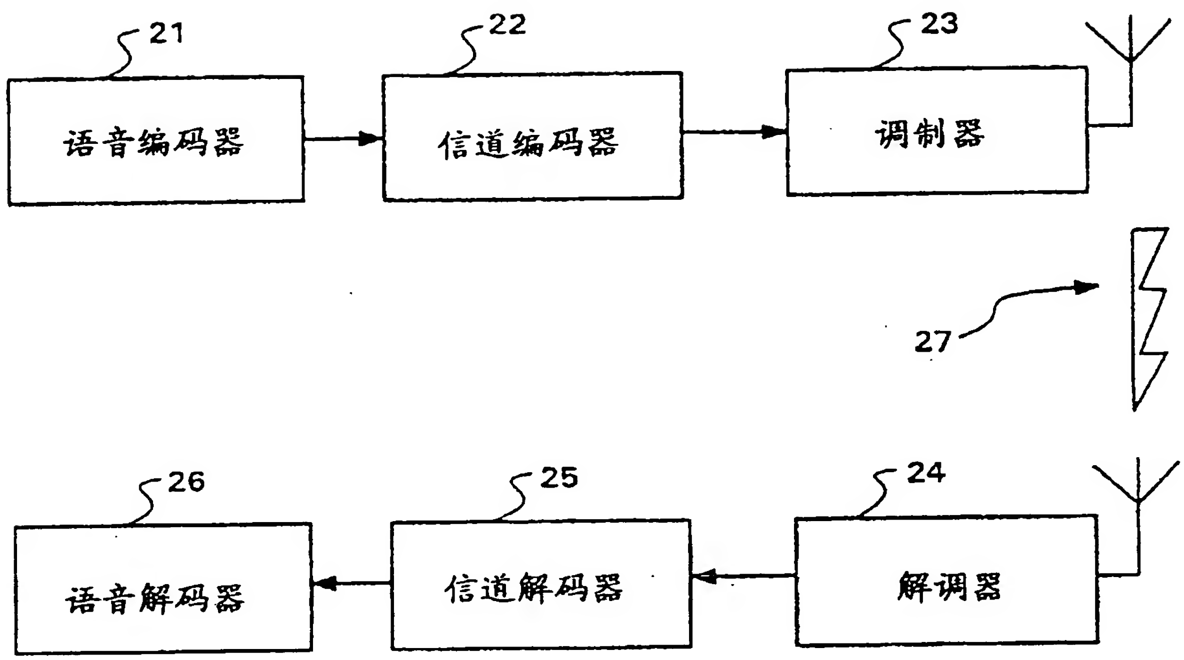


图 3

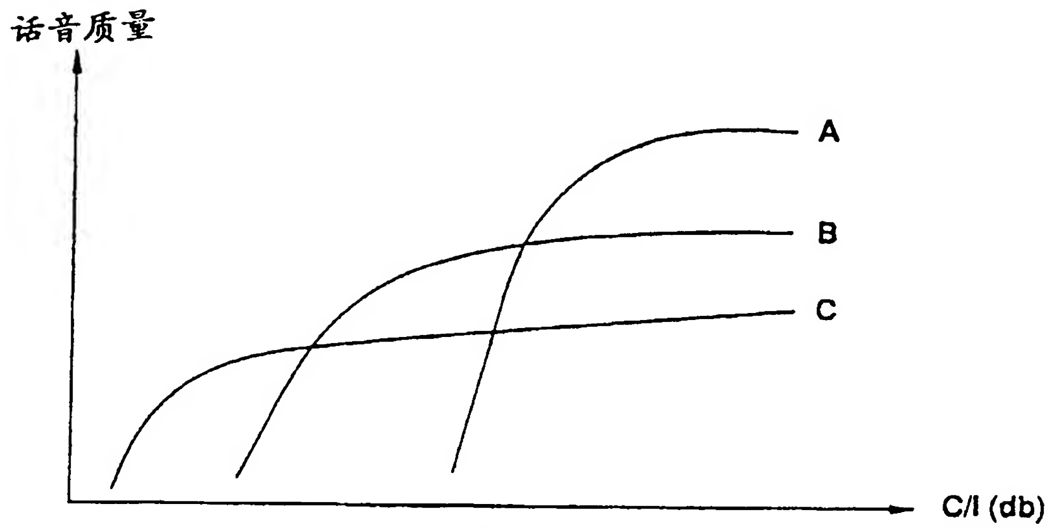


图 4

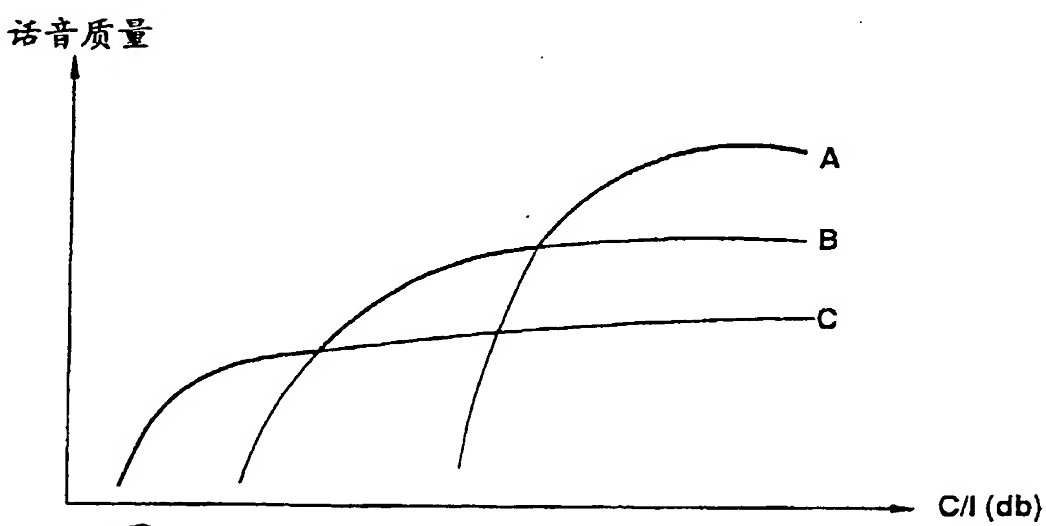


图 5

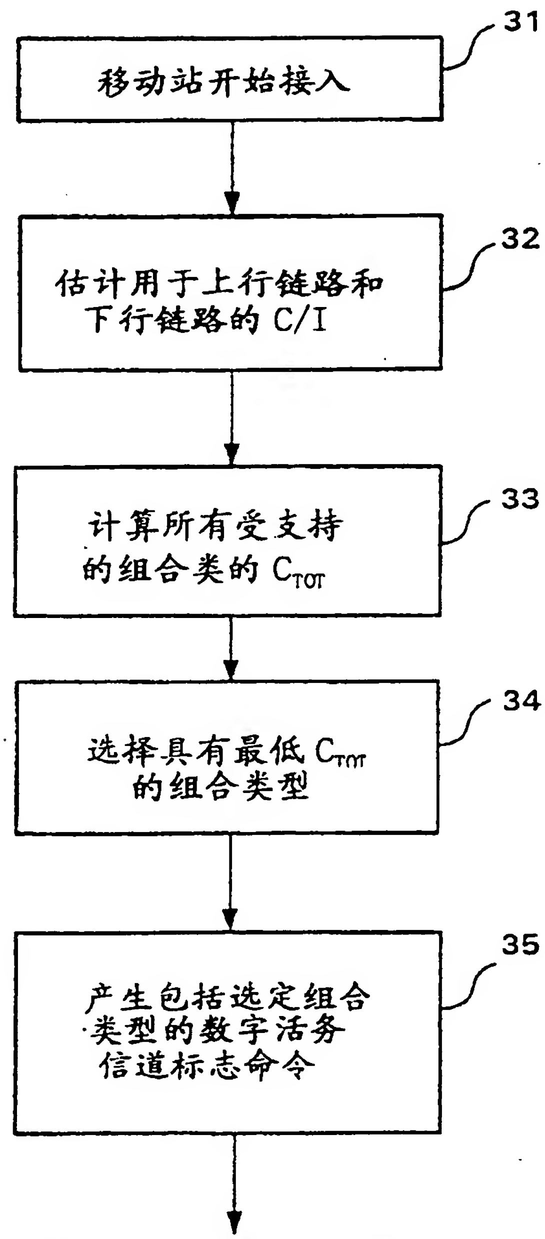

31
移动站开始接入
32
估计用于上行链路和
下行链路的 C/I
33
计算所有受支持
的组合类的 C_{TOT}
34
选择具有最低 C_{TOT}
的组合类型
35
产生包括选定组合
类型的数字活务
信道标志命令

图 6 到图 7 的步骤 41

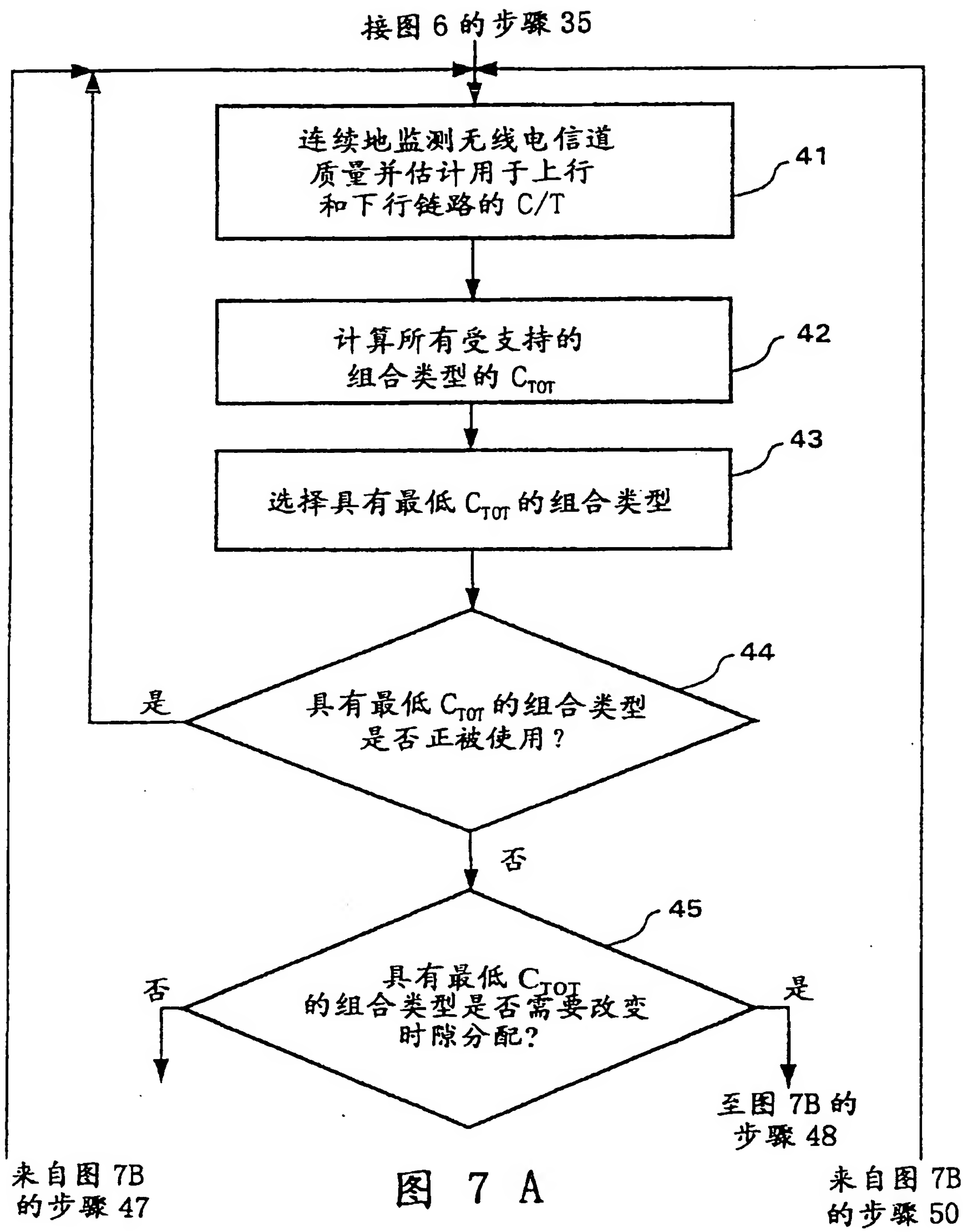

接图 6 的步骤 35
连续地监测无线电信道质量并估计用于上行和下行链路的 C/T
41
计算所有受支持的组合类型的 C_{TOT}
42
选择具有最低 C_{TOT} 的组合类型
43
具有最低 C_{TOT} 的组合类型是否正被使用？
44
是
否
具有最低 C_{TOT} 的组合类型是否需要改变时隙分配?
45
否
是
至图 7B 的步骤 48
来自图 7B 的步骤 47
来自图 7B 的步骤 50

图 7 A

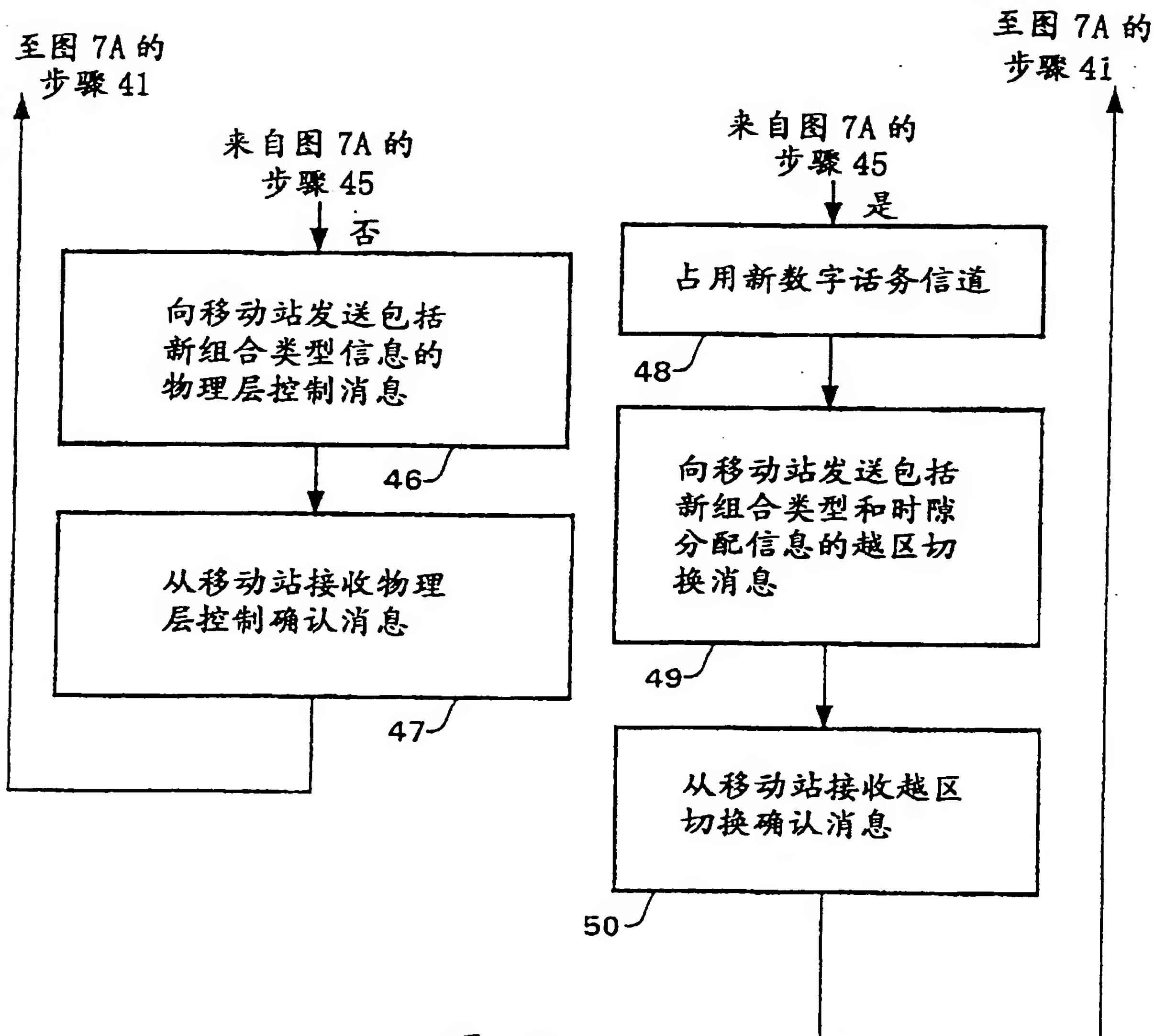

至图 7A 的
步骤 41
来自图 7A 的
步骤 45
否
向移动站发送包括
新组合类型信息的
物理层控制消息
46
从移动站接收物理
层控制确认消息
47
至图 7A 的
步骤 41
来自图 7A 的
步骤 45
是
占用新数字话务信道
48
向移动站发送包括
新组合类型和时隙
分配信息的越区切
换消息
49
从移动站接收越区
切换确认消息
50

图 7B

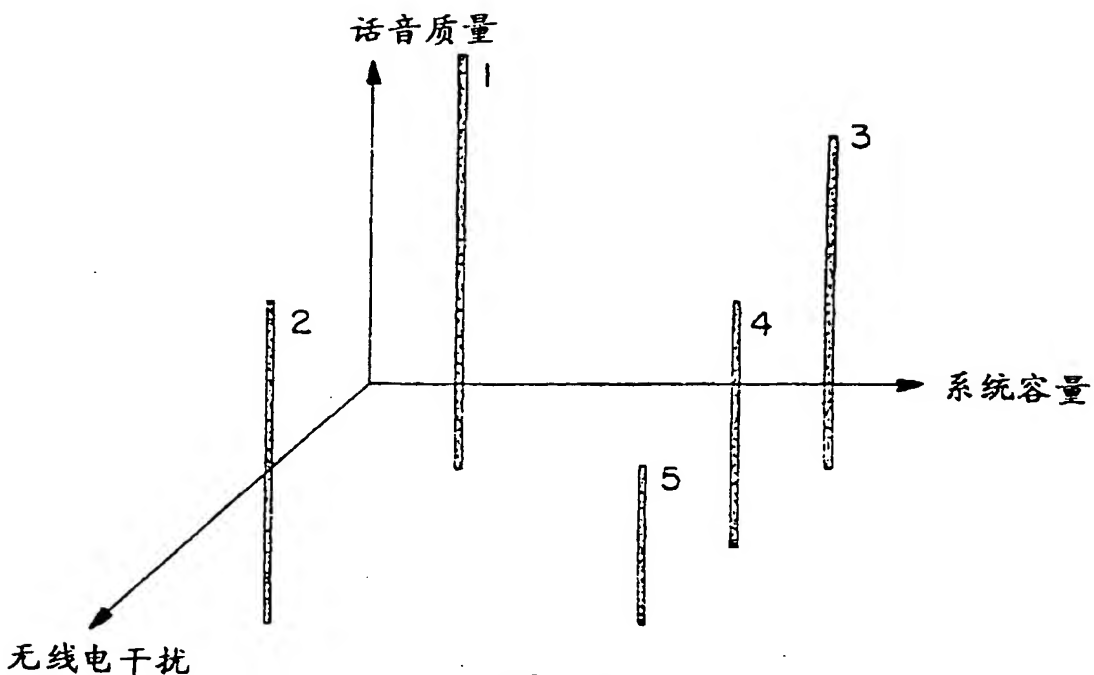


图 8

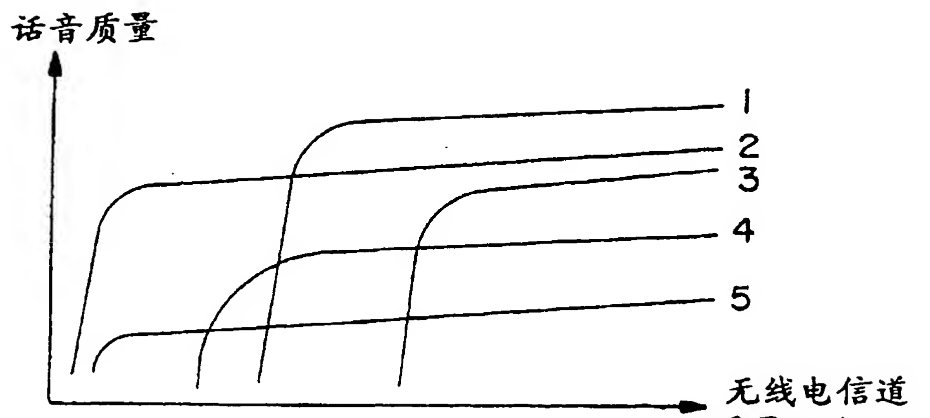


图 9

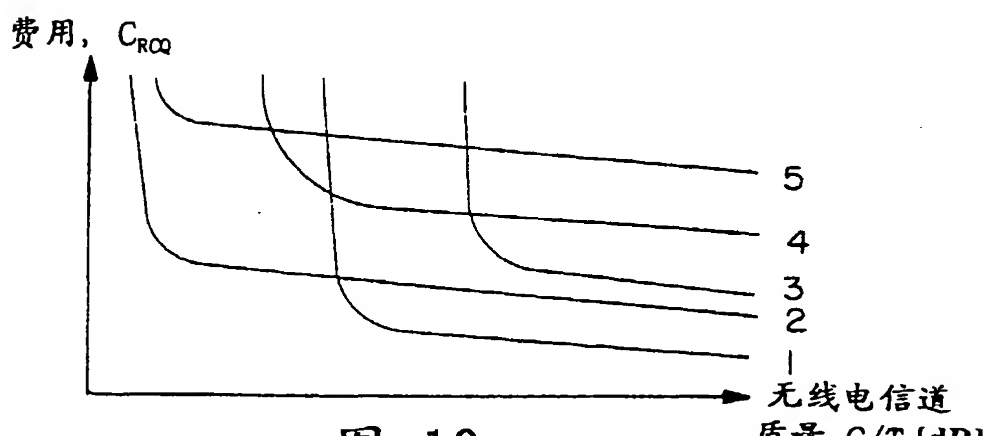


图 10

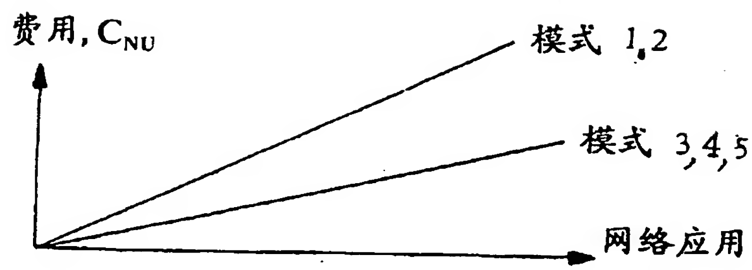


图 11

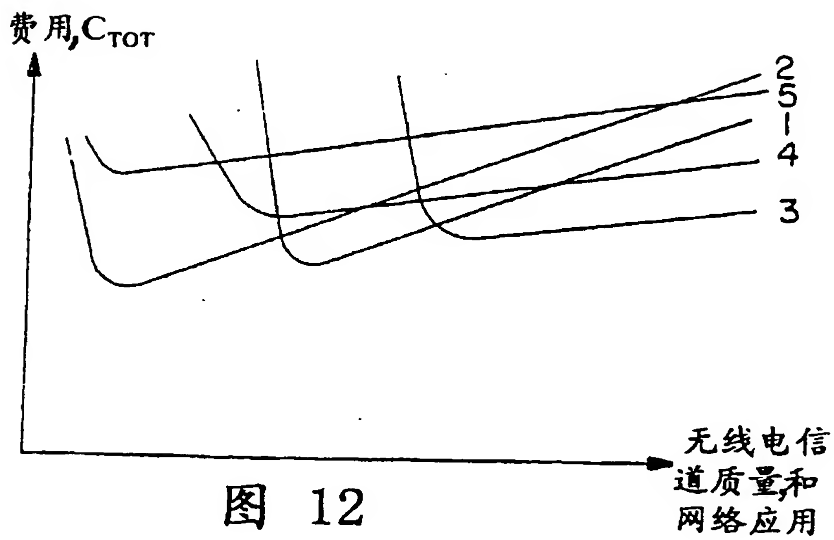


图 12

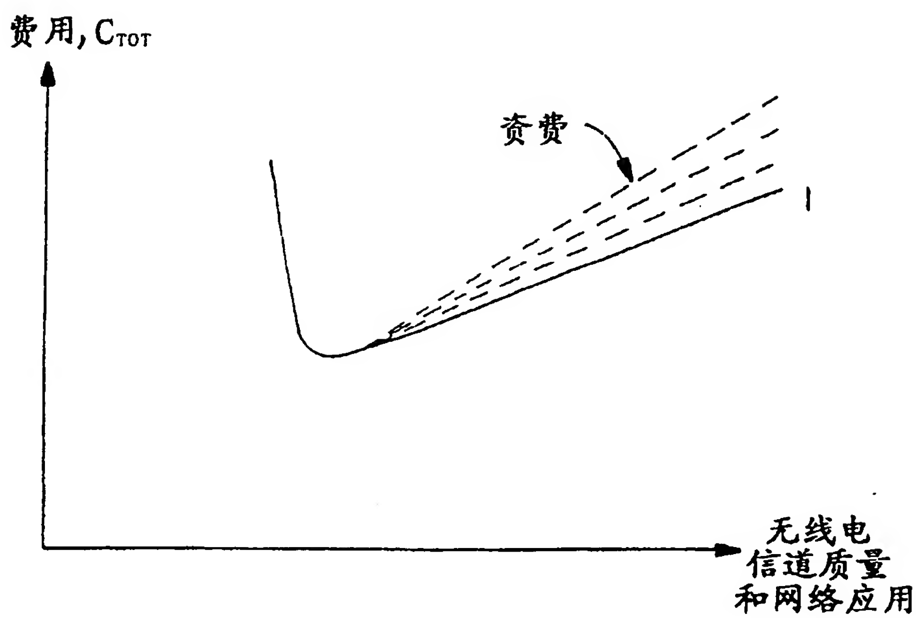


图 13

CN 1203009A

SYSTEM AND METHOD FOR FLEXIBLE CODING, MODULATION, AND TIME SLOT ALLOCATION IN A RADIO TELECOMMUNICATIONS NETWORK

BACKGROUND OF THE INVENTION

Technical Field of the Invention

This invention relates to radio telecommunication systems and, more particularly, to a system and method for improving voice quality and radio channel quality through flexible coding, modulation, and time slot allocation.

Description of Related Art

In modern cellular telecommunication systems, the geographic area of coverage may be divided into a plurality of continuous radio coverage areas, or cells, each of which is served by one base station. Each of the base stations includes a transmitter, receiver, and a base station controller as are well known in the art. A Mobile Switching Center (MSC) is connected by communication links to each of the base stations and to the Public Switched Telephone Network (PSTN) or a similar fixed network which may be include an Integrated Services Digital Network (ISDN) facility. Similarly, it is also known to include more than one MSC in the cellular radio system and to connect each additional MSC to a different group of base stations and to other MSCs via cables or radio links. A mobile station may roam freely about the service area. As mobile stations roam about the service area of the system, they are handed off from one cell to another so that there is no lapse in service.

Each of the cells is allocated a plurality of voice or speech channels and at least one access or control channel. The control channel is used to control or supervise the operation of the mobile terminal by means of information transmitted and received from those units, referred to as messages. Control and administration messages within a cellular radio system are sent in accordance with industry established air interface standards, such as AMPS and EIA/TIA 553, the standards for analog cellular operations, and/or D-AMPS, EIA/TIA 627, and TIA IS-136, the standards for digital cellular operations, all of which are hereby incorporated by reference herein. While

these standards govern North American operations, similar standards govern other geographic areas throughout the world, and are well known to those skilled in the art.

The information exchanged between base stations and mobile terminals via messages, may include incoming call signals, outgoing call signals, paging signals, paging response signals, location registration signals, voice channel assignments, maintenance instructions and handoff instructions as the mobile terminals travel out of the radio coverage of one cell and into the radio coverage of other cells, as well as other additional items of information such as calling party numbers, time information, and the like. The control or voice channels may operate in either analog or digital mode or a combination thereof based upon industry standards. Integrated services between different cellular telecommunication systems and different MSCs are provided by using the intersystem specification IS-41, which is hereby incorporated by reference herein.

The growing number of mobile stations in use today has generated the need for more voice channels within cellular telecommunication systems. Base stations have become more closely spaced, with an increase in interference between mobile stations operating on the same frequency in neighboring or closely spaced cells. Additionally, since the frequency spectrum allocated to cellular telecommunications is finite, this has lead to more closely spaced channel frequencies along with an attendant increase in interference from other channels. Digital techniques such as time division multiplexing and code division multiplexing of signals have been developed in order to gain more useful channels from a given frequency spectrum.

There still remains a need to reduce interference, or more specifically, to increase the ratio of the carrier signal strength to interference strength, (i.e., carrier-to-interference (C/I) ratio). As used herein, C/I is defined as the total carrier-to-interference ratio, where interference comprises interference from other mobile stations as well as noise (both receiver-generated and thermal).

In cellular radio systems, the user bit rate is a finite resource. For a system with a given user bit rate, there is a trade-off between voice quality in error free conditions (high C/I ratio) and the system's robustness against poor radio channel quality (low C/I ratio). A system may give priority to either voice quality or

robustness at the expense of the other characteristic. For example, a system with a user bit rate giving priority to voice quality in error-free conditions performs well at high C/I levels, but is less resistant to low C/I levels than a system with a user bit rate that gives priority to robustness. In other words, the voice quality of the first system deteriorates more rapidly as C/I levels decrease. Likewise, a system with a user bit rate giving priority to robustness is more resistant to low C/I levels, but does not perform as well at high C/I levels as a system with a user bit rate that gives priority to voice quality. In other words, the voice quality of the robust system deteriorates less rapidly as C/I levels decrease, but does not have as good a voice quality in good radio conditions.

The total user bit rate in a cellular radio system is determined by a selected combination of techniques for speech coding, channel coding, modulation, and for a time division multiple access (TDMA) system, the number of assignable time slots per call specified in the Air Interface standard. A fixed combination of the above techniques is defined by air interface standards such as IS-136. There are drawbacks, however, to having this combination specified because of constraints on achievable voice quality that arise from using a specified combination in inappropriate radio channel quality conditions. Each specified combination is optimized for a specific level of radio channel quality (C/I ratio), thereby sacrificing voice quality when the C/I ratio is high, and/or sacrificing robustness when the C/I ratio is low. The cellular air interface standards of today specify fixed combinations that either provide high voice quality in high C/I conditions, or provide robustness. The combination for high voice quality in high C/I conditions produces a system that is less robust, and the voice quality is unacceptably poor in low C/I conditions. The combination for robustness sacrifices high voice quality in high C/I conditions in exchange for acceptable voice quality in low C/I conditions, even though the robustness may be needed in only a limited number of cases when C/I conditions are low.

Although there are no known prior art teachings of a solution to the aforementioned deficiency and shortcoming such as that disclosed herein, a number of prior art references exist that discuss subject matter that bears some relation to matters discussed herein. Such prior art references are U.S. Patent Number 5,134,615

to Freeburg et al., and an IEEE article by J. Woodard and L. Hanzo entitled, "A Dual-rate Algebraic CELP-based Speech Transceiver". Each of these references is discussed briefly below.

U.S. Patent Number 5,134,615 to Freeburg et al. (Freeburg) discloses a method of selecting frequency and time slot assignments for communication with devices having different communication protocols, including different available time slots. An adaptable time slot selector is included, allowing communication with devices using other protocols. Freeburg, however, only addresses time slot allocation in the context of providing communications with devices utilizing different air interface protocols. Freeburg does not in any way teach or suggest a method of achieving improved voice quality in a digital cellular system over a broad range of C/I conditions. The present invention dynamically adapts a cellular system's combination of speech coding, channel coding, modulation, and number of assignable time slots per call to achieve the optimum voice quality for the currently measured C/I conditions.

The IEEE article by J. Woodard and L. Hanzo entitled, "A Dual-rate Algebraic CELP-based Speech Transceiver" (Woodard) discloses a system that utilizes two combinations of speech coding, channel coding, and modulation called Low-quality mode and High-quality mode. Woodard, however, does not teach or suggest any process for quality-driven or capacity-driven selection of the different modes. Woodard does not in any way suggest a system that dynamically adapts a cellular system's combination of speech coding, channel coding, modulation, and number of assignable time slots per call to achieve optimum voice quality over a broad range of C/I conditions.

Review of each of the foregoing references reveals no disclosure or suggestion of a system or method such as that described and claimed herein.

It would be a distinct advantage to have a system and method for dynamically adapting a cellular system's combination of speech coding, channel coding, modulation, and number of assignable time slots per call to achieve optimum voice quality over a broad range of C/I conditions. The present invention provides such a system and method.

SUMMARY OF THE INVENTION

The present invention is a system and method for dynamically adapting the user bit rate of a time division multiple access (TDMA) cellular telecommunication system to achieve optimum voice quality over a broad range of radio channel conditions. The system continuously monitors radio channel quality both on an uplink (from a mobile station to its serving base station) and on a downlink (from the serving base station to the mobile station), and dynamically adapts the system's combination of speech coding, channel coding, modulation, and number of assignable time slots per call to optimize voice quality for the measured conditions. Various combinations of the system's speech coding, channel coding, modulation, and assignable time slots are identified as combination types. In addition, cost functions may be introduced, and by identifying and selecting the cost function with the lowest cost for the measured radio channel conditions, the system provides the maximum voice quality achievable within the limits of the system design.

In another aspect, the present invention is a system for dynamically optimizing voice quality in a digital cellular radio telecommunications network having a plurality of user bit rate components that operate at set bit rates. The network utilizes a plurality of radio channels to carry calls. The system comprises means for monitoring and measuring conditions on each of the radio channels,
means for estimating current radio channel quality for each of the radio channels, means for changing the bit rates of each of the plurality of user bit rate components, and means for dynamically controlling the means for changing bit rates in order to provide the maximum achievable voice quality for calls on each of the radio channels.

In still another aspect, the present invention is a method of dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network having a plurality of user bit rate components that operate at set bit rates. The network utilizes a plurality of time slots to carry a plurality of calls on each radio channel. The method begins by monitoring and measuring conditions on each of the radio channels, and estimating current radio channel quality for each of the radio channels. The method then dynamically changes the bit rates and

allocates time slots based upon the estimated radio channel quality, thereby providing the maximum achievable voice quality for calls on each of the radio channels.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be better understood and its numerous objects and advantages will become more apparent to those skilled in the art by reference to the following drawing, in conjunction with the accompanying specification, in which:

FIG. 1 (Prior art) is a graph of mobile station signal strength at a serving base station as a function of mobile station distance from the base station;

FIG. 2 (Prior art) is a graph of an exemplary cumulative distribution function illustrating the percentage of mobile stations (MSs) in an illustrative cellular telecommunication system that are, at a given time, experiencing a ratio of carrier signal strength to interference (C/I) below corresponding levels;

FIG. 3 (Prior art) is a simplified block diagram of the components in a base station and a mobile station in a cellular telecommunication system that contribute to the total user bit rate;

FIG. 4 (Prior art) is a graph of voice quality as a function of carrier signal-to-interference ratio (C/I) for three exemplary combinations of the components of FIG. 3;

FIG. 5 is a graph of voice quality as a function of carrier signal-to-interference ratio (C/I) illustrating the effect on voice quality provided by adaptive selection of the optimum combination of the components of FIG.3 at varying levels of C/I ratio;

FIG. 6 is a flow chart illustrating the functions performed by the control program of the present invention when dynamically selecting a combination type during call setup;

FIGS. 7A and 7B are a flow chart illustrating the functions performed by the control program of the present invention when dynamically selecting a combination type during a call in progress;

FIG. 8 is a 3-dimensional graphical presentation illustrating the achievable voice quality levels of the five exemplary combination types in varying conditions of radio interference and cellular system capacity;

FIG. 9 is a graph of voice quality as a function of radio channel quality (RCQ) or C/I for the five exemplary combination types of Table II;

FIG. 10 is a graph of cost versus C/I ratio utilized by the control program of the present invention and illustrating curves for different tariffs;

FIG. 10 is a graph of Cost (C_{RCQ}) as a function of radio channel quality (RCQ) or C/I for the five exemplary combination types of Table II;

FIG. 11 is a graph of Cost (C_{SU}) as a function of system utilization (SU);

FIG. 12 is a graph of Total Cost (C_{TOT}) as a function of both radio channel quality and system utilization; and

FIG. 13 is a graph of the total cost (C_{TOT})curve for combination type 1 and illustrating the application of different "tariffs" to the total cost curve.

DETAILED DESCRIPTION OF EMBODIMENTS

FIG. 1 is a graph of mobile station signal strength at a serving base station as a function of mobile station distance from the base station. Current systems measure the signal strength of mobile stations at the base station, and as a mobile station travels away from its serving base station, the measured signal strength decreases. When the signal strength reaches a minimum acceptable level 11, the output power of the mobile station is increased incrementally, thereby increasing the signal strength received at the base station to an intermediate level 12. The intermediate level 12 is intended to provide acceptable signal strength and acceptable interference levels to other mobile stations. As the mobile station continues to increase its distance from the base station, the signal strength again decreases to the minimum acceptable level, and the output power of the mobile station is again incrementally increased. This process continues until the mobile station is operating at its maximum output capability. If the signal strength then falls to the minimum acceptable level, the mobile station is handed off to another cell, if possible, or the call is dropped.

FIG. 2 is a graph of an exemplary cumulative distribution function illustrating the percentage of mobile stations (MSs) in an illustrative cellular telecommunication system that are, at a given time, experiencing a ratio of carrier signal strength to interference (C/I) below corresponding levels. FIG. 2, which is exemplary only,

illustrates, for example, that 100% of the mobile stations are experiencing C/I ratios of 50 dB or lower. Approximately 80% are experiencing C/I ratios of 30 dB or lower. Likewise, almost none of the mobile stations are experiencing a C/I ratio so low that a call cannot be maintained. The C/I ratio is the measurement generally recognized as being indicative of the voice quality on a given cellular radio channel, with higher ratios providing better voice quality. The C/I measurement actually includes a noise factor as well, but in interference-limited environments, the effect of noise on voice quality is negligible when compared to the effect of interference.

FIG. 3 is a simplified block diagram of the components in a base station and a mobile station in a cellular telecommunication system that contribute to the achievable voice quality. In existing cellular telecommunication systems, the gross bit rate is determined by a combination of the bit rates of a speech coder 21, channel coder 22, and modulator 23 when transmitting; a demodulator 24, channel decoder 25, and speech decoder 26 when receiving; and, for a time division multiple access (TDMA) system, the number of assignable time slots per call in the Air Interface 27. Allowable bit rates for each of the above components are specified by existing telecommunications standards. A large number of possible combinations of bit rates exist, and may be chosen by an operator in order to prioritize either voice quality or robustness.

FIG. 4 is a graph of voice quality as a function of carrier-to-interference ratio (C/I) for three exemplary combinations of the components of FIG. 3. While only three combinations have been illustrated for simplicity, it should be understood that this is exemplary only, and in practice many more may be utilized.

Combination types A, B, and C are represented as curves illustrating the voice quality attainable for each combination for varying levels of C/I ratio. Combination type A offers the best voice quality of the three user bit rates, but is the least robust and is only useful at the higher levels of C/I. With combination type A, voice quality rapidly deteriorates to an unacceptable level as C/I decreases. Combination type C, the other extreme, is the most robust. Therefore, as C/I decreases, voice quality under combination type C deteriorates very slowly and combination type C provides the best voice quality at low levels of C/I. However, combination type C sacrifices

voice quality at high levels of C/I where its achievable voice quality is the lowest of the three exemplary combinations.

Combination type B offers a compromise between the good voice quality performance of combination type A in areas of high C/I and the robustness of combination type C. Combination type B may offer higher voice quality in the middle ranges of C/I than either combination type A or C. At high levels of C/I, combination type B offers higher voice quality than combination type C, but lower voice quality than combination type A. In areas of low C/I, combination type B offers higher voice quality than combination type A, but lower voice quality than combination type C. Combinations similar to combination type B are most often utilized by cellular air interface standards since those combinations offer medium performance throughout most of the C/I range normally experienced.

FIG. 5 is a graph of voice quality as a function of carrier signal-to-interference ratio (C/I) illustrating the effect on voice quality provided by adaptive selection of the optimum combination of the components of FIG.3 at varying levels of C/I ratio. Adaptive selection provides a cellular radio system with achievable voice quality illustrated by the bold curve in FIG. 5. In the system of the present invention, the instantaneous radio channel quality (i.e., C/I ratio) is continuously monitored. The voice quality and required level of robustness for the measured C/I ratio are known for each of the combination types A, B, and C discussed above. The system dynamically responds to the measured C/I by selecting whichever of the combination types A, B, or C gives the maximum voice quality for the required robustness at the measured C/I level. Thus, using the exemplary curves of FIGs. 4 and 5, the system utilizes combination type A at high levels of C/I, combination type B at intermediate levels of C/I, and combination type C at low levels of C/I. Therefore, voice quality is dynamically maximized.

The present invention includes a control algorithm that selects the best combination for a given level of C/I. The control algorithm is based on "cost functions" which provide for simple and stable decision making. The switching between different combinations may be controlled by either the cellular system or by the mobile stations.

One of the current TDMA standards in North America (IS-136) specifies a three time slot structure, i.e., every third time slot is allocated to one particular user. The voice quality curves for FIGS. 4 and 5 are illustrative of the achievable voice quality when utilizing a single time slot out of three for each user. Additional time slots may be allocated to a single user, but such an allocation adversely impacts the capacity of the system by decreasing the number of users per frequency. However, allocating additional time slots to a user implicitly increases bandwidth to that particular user and improves voice quality. It may be desirable, therefore, to allocate additional time slots to each user during periods of low traffic density when system capacity is not a problem. Therefore, the control program of the present invention is enhanced by using "tariffs" (sets of cost functions), depending on the current load and the category of subscriber. This provides the cellular system operator with the ability to trade off voice quality for system capacity, or provide additional bandwidth to subscribers who are willing to pay a premium for additional capabilities.

FIG. 6 is a flow chart illustrating the functions performed by the control program of the present invention when dynamically selecting a combination type during call setup. The program starts at step 31 where the mobile station originates access to the cellular telecommunications network. The program then moves to step 32 and estimates the radio channel quality (RCQ) (e.g., C/I) based on idle channel measurements, for both the uplink signal (from the mobile station to the base station) and the downlink signal (from the base station to the mobile station). At step 33, the program calculates the total cost (C_{TOT}) for all user bit rate combination types that are supported by both the cellular network and the mobile station. At step 34, the program selects the combination type with the lowest total cost to utilize for the uplink and the downlink transmissions. The program then generates a Digital Traffic Channel (DTC) designation order at step 35 which includes information concerning the combination type to utilize for the uplink and the downlink transmissions. The program then moves to step 41 in FIG. 7.

FIGS. 7A and 7B are a flow chart illustrating the functions performed by the control program of the present invention when dynamically selecting a combination type during a call in progress. At step 41, the program continuously monitors the radio

channel and estimates radio channel quality (C/I) for both the uplink and the downlink. At step 42, the program calculates the total cost (C_{TOT}) for all user bit rate combination types that are supported by both the cellular network and the mobile station. At step 43, the program selects the combination type with the lowest total cost to utilize for the uplink and the downlink transmissions. At step 44, it is determined whether or not the combination type with the lowest C_{TOT} is currently being utilized in the uplink and the downlink. If it is determined that the combination type with the lowest C_{TOT} is currently being utilized in the uplink and the downlink, then no action is taken, and the program returns to step 41 and continues to monitor the radio channel and estimate radio channel quality.

If at step 44, however, it is determined that the combination type with the lowest C_{TOT} is not currently being utilized in the uplink and the downlink, then the program performs the functions required to change the uplink and the downlink to a new user bit rate combination type. To accomplish this, the program first moves to step 45 and determines whether or not the combination type with the lowest C_{TOT} requires a change in time slot allocation. If no change in time slot allocation is required, then the program moves to FIG. 7B, step 46 and sends to the mobile station a Physical Layer Control message which includes information about the new combination type to utilize in the uplink and the downlink. The switch to the new combination type is complete when the program receives a Physical Layer Control Acknowledgement message at step 47. The program then returns to FIG. 7A, step 41 and continues to monitor the radio channel and estimate radio channel quality.

If at step 45, however, it is determined that the combination type with the lowest C_{TOT} requires a change in time slot allocation, then a handoff is initiated. The program first moves to FIG. 7B, step 48 and seizes a new digital traffic channel. Then program then moves to step 49 and sends to the mobile station a Handoff message which includes information about the new combination type (including new time slot allocation) to utilize in the uplink and the downlink. The switch to the new combination type is complete when the program receives a Handoff Acknowledgement message at step 50. The program then returns to FIG. 7A, step 41 and continues to monitor the radio channel and estimate radio channel quality.

The control program continuously monitors and measures radio channel conditions which determine radio channel quality (RCQ) for both the uplink and the downlink, as well as other cellular network conditions which may influence achievable voice quality. These conditions may include, for example:

<u>Radio Channel Conditions</u>:
Bit Error Rate (BER) - uplink;
Bit Error Rate (BER) - downlink;
Signal Strength (SS) - uplink; and
Signal Strength (SS) - downlink.

<u>Cellular Network Conditions</u>:
Available Time Slots;
Mobile Station (MS) Capability;
Cellular System Capability; and
Tariffs.

The control program monitors these conditions and, based on its measurements, optimizes the voice quality for each individual call, in order to achieve the best possible quality within the given resources (e.g., time slots, MS capability, etc.) of the cellular system. Cost functions are introduced in order to provide a flexible trade-off between system capacity and voice quality. The above measured factors are input to the control program which then applies the cost functions to select the combination type that minimizes the total cost.

The control program estimates the current radio channel quality (C/I) based on Bit Error Rate (BER) estimates (uplink and downlink) and Signal Strength (SS) estimates (uplink and downlink). The control program may utilize a look-up table similar to Table I below to translate BER to C/I.

UPLINK		DOWNLINK	
BER (%)	C/I (dB)	BER (%)	C/I (dB)
10	7	10	10
5	9	5	13
3	11	3	17

Detailed Example.

A detailed example is hereinafter described to illustrate a typical implementation of the present invention. In this example, a total of five (5) combination types are available for use in the cellular network. The five combination types are defined in Table II below:

Type	Time Slots[a]	Modulation	Voice Coder Algorithm, Rate (kbps)		Total Data Rate[b] (kbps)
1	5	8PSK	ADPCM	32	56.75
2	5	π/4 QPSK	LDCELP	16	37.8
3	2	8PSK	LDCELP	16	19.5
4[c]	2	π/4 QPSK	VSELP	7.95	13
5	2	π/4 QPSK	EVCELP[d]	4.0	13

[a] Number of IS-136 air interface time slots per 40 ms (out of 6)
[b] Including voice coder rate and Forward Error Correction (FEC) coding
[c] IS-136 full rate
[d] Enhanced VSELP

The following glossary expands the terms utilized in the table:

PSK	Phase Shift Keying
QPSK	Quadrature Phase Shift Keying
ADPCM	Adaptive Differential Pulse Code Modulation
LDCELP	Low Delay Code Excited Linear Predictive coding
VSELP	Vector Sum Excited Linear Predictive coding

Combination types 1 and 2 are optimized for indoor/office applications where the system capacity per frequency is less of a problem because, for example, microcells may be implemented in order to obtain the required capacity. Combination types 3, 4, and 5 are optimized for outdoor/wide area applications where maximum capacity per cell/frequency is required.

FIG. 8 is a 3-dimensional graphical presentation illustrating the achievable voice quality levels of the five exemplary combination types in varying conditions of radio interference and cellular system capacity. Combination types 1 and 3 provide

very good voice quality in good radio conditions (high C/I ratio) and at different levels of cellular system capacity. Combination types 2, 4, and 5 provide optimal voice quality in various levels of degraded radio conditions (lower C/I ratios) and at different levels of cellular system capacity.

FIG. 9 is a graph of voice quality as a function of radio channel quality (RCQ) or C/I for the five exemplary combination types of Table II. Each curve represents one of the combination types, as indicated. It can be seen from FIG. 9 that combination types 2 and 5 are the most robust, offering degraded, but acceptable voice quality at the lower levels of C/I. Curves 1 and 3, conversely, offer good voice quality at high levels of C/I, but degrade rapidly to unacceptable voice quality at lower levels of C/I.

FIG. 10 is a graph of Cost (C_{RCQ}) as a function of radio channel quality (RCQ) or C/I for the five exemplary combination types of Table II. Each curve represents one of the combination types, as indicated. The curves illustrate that it is more costly to achieve acceptable voice quality at lower levels of C/I than it is to achieve acceptable voice quality at higher levels of C/I. This fact results in cost curves for the five combination types that are essentially the inverse of the voice quality curves of FIG. 9. Therefore, by selecting the lowest cost curve for a given radio channel quality (C/I), the combination type providing the best voice quality is also chosen.

FIG. 11 is a graph of Cost (C_{NU}) as a function of cellular network utilization (NU). The curves represent combination types as indicated. The curves in FIG. 11 illustrate that it is more costly to allocate additional time slots to each user during periods of high system utilization. This occurs because there are fewer time slots available for increasing voice quality due to their utilization to serve more users.

FIG. 12 is a graph of Total Cost (C_{TOT}) as a function of both radio channel quality and system utilization. The total cost (C_{TOT}) for each combination type is the sum of C_{RCQ} and C_{NU} for that combination. As noted above, the control program continuously monitors radio channel quality and system utilization, and selects the lowest total cost curve. This results in the selection of the combination type providing the best voice quality within the constraints of the cellular system resources.

FIG. 13 is a graph of the total cost (C_{TOT}) curve for combination type 1 and illustrating the application of different "tariffs" to the total cost curve. Tariffs provide

the cellular system operator with the ability to tailor system usage to groups of subscribers with different priority levels. Achieving this level of voice quality may require access to a combination type that utilizes additional time slots. By levying tariffs on the total cost function, the system operator can offer this service to those subscribers who are willing to pay increased charges to obtain the additional time slots. The operator can also control the size of the tariffs, thereby having greater tariffs during periods of high system utilization when available time slots are scarce. Thus, tariffs adjust the total cost function depending on network utilization, radio channel quality, and demand for network resources.

It is thus believed that the operation and construction of the present invention will be apparent from the foregoing description. While the method, apparatus and system shown and described has been characterized as being preferred, it will be readily apparent that various changes and modifications could be made therein without departing from the spirit and scope of the invention as defined in the following claims.

WHAT IS CLAIMED IS:

1. A system for dynamically optimizing voice quality in a digital cellular radio telecommunications network, said network having a plurality of user bit rate components that operate at set bit rates, and said network utilizing a plurality of radio channels to carry calls, said system comprising:

means for monitoring and measuring conditions on each of said radio channels;

means for estimating current radio channel quality for each of said radio channels;

means for changing the bit rates of each of said plurality of user bit rate components; and

means for dynamically controlling said means for changing bit rates in order to provide the maximum achievable voice quality for calls on each of said radio channels.

2. The system for dynamically optimizing voice quality in a digital cellular radio telecommunications network of claim 1 wherein said means for monitoring and measuring conditions on each of said radio channels includes means for continuously monitoring and measuring said conditions.

3. The system for dynamically optimizing voice quality in a digital cellular radio telecommunications network of claim 2 further comprising means for monitoring and measuring cellular network conditions that influence achievable voice quality.

4. The system for dynamically optimizing voice quality in a digital cellular radio telecommunications network of claim 3 wherein said cellular network conditions that influence achievable voice quality include:

mobile station (MS) capability;

cellular network capability; and

tariffs.

5. The system for dynamically optimizing voice quality in a digital cellular radio telecommunications network of claim 3 wherein said plurality of user

bit rate components includes a speech encoder, a channel encoder, a modulator, a speech decoder, a channel decoder, and a demodulator.

6. The system for dynamically optimizing voice quality in a digital cellular radio telecommunications network of claim 5 wherein said means for continuously monitoring and measuring conditions on each of said radio channels includes means for continuously monitoring and measuring bit error rates (BER) and signal strengths (SS).

7. The system for dynamically optimizing voice quality in a digital cellular radio telecommunications network of claim 6 wherein said means for dynamically controlling said means for changing bit rates includes:

means for defining a plurality of combination types, each of said plurality of combination types comprising a defined bit rate for each of said plurality of user bit rate components;

means for defining a plurality of cost functions, each of said cost functions corresponding to one of said plurality of combination types; and

means for identifying and selecting a cost function that provides the lowest cost for said measured radio channel conditions.

8. The system for dynamically optimizing voice quality in a digital cellular radio telecommunications network of claim 7 wherein said means for defining a plurality of cost functions includes:

means for defining cost as a function of radio channel quality;

means for defining cost as a function of cellular network utilization; and

means for adding said cost as a function of radio channel quality and said cost as a function of cellular network utilization to obtain a total cost function for each of said plurality of combination types.

9. The system for dynamically optimizing voice quality in a digital cellular radio telecommunications network of claim 8 wherein said means for defining a plurality of cost functions includes means for applying tariffs to the total cost function for each of said plurality of combination types, said tariffs adjusting said total cost functions depending on network utilization, radio channel quality, and demand for network resources.

10. A system for dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network, said network having a plurality of user bit rate components that operate at set bit rates, and said network utilizing a plurality of time slots to carry a plurality of calls on each radio channel, said system comprising:

means for monitoring and measuring conditions on each of said radio channels;

means for estimating current radio channel quality for each of said radio channels;

means for changing the bit rates of each of said plurality of user bit rate components;

means for allocating time slots to selected calls; and

means for dynamically controlling said means for changing bit rates and said means for allocating time slots in order to provide the maximum achievable voice quality for calls on each of said radio channels.

11. The system for dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 10 wherein said means for monitoring and measuring conditions on each of said radio channels includes means for continuously monitoring and measuring said conditions.

12. The system for dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 11 further comprising means for monitoring and measuring cellular network conditions that influence achievable voice quality.

13. The system for dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 12 wherein said cellular network conditions that influence achievable voice quality include:

available time slots;

mobile station (MS) capability;

cellular network capability; and

tariffs.

14. The system for dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 12 wherein said plurality of user bit rate components includes a speech encoder, a channel encoder, a modulator, a speech decoder, a channel decoder, and a demodulator.

15. The system for dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 14 wherein said means for continuously monitoring and measuring conditions on each of said radio channels includes means for continuously monitoring and measuring bit error rates (BER) and signal strengths (SS).

16. The system for dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 15 wherein said means for dynamically controlling said means for changing bit rates and said means for allocating additional time slots includes:

means for defining a plurality of combination types, each of said plurality of combination types comprising:

a bit rate setting for each of said plurality of user bit rate components; and

an allocation of time slots for each call;

means for defining a plurality of cost functions, each of said cost functions corresponding to one of said plurality of combination types; and

means for identifying and selecting a cost function that provides the lowest cost for said measured radio channel conditions.

17. The system for dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 16 wherein said means for defining a plurality of cost functions includes:

means for defining cost as a function of radio channel quality;

means for defining cost as a function of cellular network utilization; and

means for adding said cost as a function of radio channel quality and said cost as a function of cellular network utilization to obtain a total cost function for each of said plurality of combination types.

18. The system for dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 17 wherein said means for defining a plurality of cost functions includes means for applying tariffs to the total cost function for each of said plurality of combination types, said tariffs adjusting said total cost functions depending on network utilization, radio channel quality, and demand for network resources.

19. A method of dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network, said network having a plurality of user bit rate components that operate at independently set bit rates, and said network utilizing a plurality of time slots to carry a plurality of calls on each radio channel, said method comprising the steps of:

monitoring and measuring conditions on each of said radio channels;

estimating current radio channel quality for each of said radio channels; and

dynamically changing said bit rates and allocating time slots, thereby providing the maximum achievable voice quality for calls on each of said radio channels.

20. The method of dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 19 wherein said step of monitoring and measuring conditions on each of said radio channels includes continuously monitoring and measuring said conditions.

21. The method of dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 20 further comprising the step of monitoring and measuring cellular network conditions that influence achievable voice quality.

22. The method of dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 21 wherein said step of monitoring and measuring cellular network conditions includes monitoring and measuring available time slots, mobile station (MS) capability, cellular network capability, and tariffs.

23. The method of dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 21 wherein said step of dynamically changing the bit rates of each of said plurality of user

bit rate components includes changing the bit rates of a speech encoder, a channel encoder, a modulator, a speech decoder, a channel decoder, and a demodulator.

24. The method of dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 23 wherein said step of continuously monitoring and measuring conditions on each of said radio channels includes continuously monitoring and measuring bit error rates (BER) and signal strengths (SS).

25. The method of dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 24 wherein said step of dynamically changing bit rates and allocating time slots includes the steps of:

defining a plurality of combination types, said defining step further comprising:

setting a bit rate for each of said plurality of user bit rate components; and

allocating a number of time slots for each call;

defining a plurality of cost functions, each of said cost functions corresponding to one of said plurality of combination types; and

identifying and selecting a cost function that provides the lowest cost for said measured radio channel conditions.

26. The method of dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 25 wherein said step of defining a plurality of cost functions includes:

defining cost as a function of radio channel quality;

defining cost as a function of cellular network utilization; and

adding said cost as a function of radio channel quality and said cost as a function of cellular network utilization to obtain a total cost function for each of said plurality of combination types.

27. The method of dynamically optimizing voice quality in a time division multiple access (TDMA) cellular radio telecommunications network of claim 26 wherein said step of defining a plurality of cost functions includes applying tariffs to

the total cost function for each of said plurality of combination types, said tariffs adjusting said total cost functions depending on network utilization, radio channel quality, and demand for network resources.

ABSTRACT OF THE DISCLOSURE

A system and method for dynamically adapting the user bit rate of a time division multiple access (TDMA) cellular telecommunication system to achieve optimum voice quality over a broad range of radio channel conditions are disclosed. The system continuously monitors radio channel quality on both the uplink and the downlink, and dynamically adapts the system's combination of speech coding, (21) channel coding (22), modulation (23), and number of assignable time slots per call (27) to optimize voice quality for the measured conditions. Various combinations of the system's speech coding, channel coding, modulation, and assignable time slots are identified as combination types (1-5) and corresponding cost functions are defined. By identifying and selecting the cost function with the lowest cost for the measured radio channel conditions, the system provides the maximum voice quality achievable within the limits of the system design.